CLAIMS

1. Apparatus for amplifying an input signal having an input signal power, comprising:

a power amplifier, which is adapted to amplify an intermediate signal so as to generate an output signal, and which is characterized by a non-linearity; and

a non-linear filter, which is coupled to decompose the input signal into a series of input signal components, each such component proportional to a product of the input signal by a respective integer power of the input signal power, and which is adapted to filter the signal components responsively to the non-linearity so as to generate the intermediate signal for input to the power amplifier.

2. The apparatus according to claim 1, wherein the non-linear filter is adapted to combine the input signal components so as to generate substantially uncorrelated input signal elements, and to filter each of the substantially uncorrelated input signal elements so as to compensate for the non-linearity.

3. The apparatus according to claim 2, wherein the non-linear filter is adapted to generate the substantially uncorrelated input signal elements by linear combination of the input signal components, using weighting coefficients determined so as to minimize a correlation between the signal elements.

4. The apparatus according to claim 2, and comprising an adaptation circuit, which is arranged to decompose the output signal so as to generate substantially uncorrelated output signal components, and to process the substantially uncorrelated output signal components in

order to determine filter coefficients to be applied by the non-linear filter in filtering the substantially uncorrelated input signal elements.

5. The apparatus according to claim 4, wherein the adaptation circuit is arranged to filter the substantially uncorrelated output signal components, and to compare the filtered substantially uncorrelated output signal components to the intermediate signal in order to determine the filter coefficients to be applied by the non-linear filter to each of the substantially uncorrelated input signal components.

6. The apparatus according to claim 5, wherein the adaptation circuit is arranged to compute successive differences between the intermediate signal and the filtered substantially uncorrelated output signal components, in order to determine a respective error signal for use in adaptively computing the filter coefficients to be applied to each of the substantially uncorrelated input signal components.

7. The apparatus according to claim 4, wherein the non-linear filter is adapted to generate the substantially uncorrelated input signal elements by linear combination of the input signal components, and wherein the adaptation circuit is arranged to adaptively determine weighting coefficients to be applied by the non-linear filter in generating the substantially uncorrelated input signal elements and to be applied by the adaptation circuit in generating the substantially uncorrelated output signal elements.

8. The apparatus according to claim 2, wherein the non-linear filter is adapted to sum the filtered

substantially uncorrelated input signal elements in order to generate the intermediate signal.

9. The apparatus according to claim 1, and comprising an adaptation circuit, which is coupled to receive and process samples of the output signal so as to produce an inverse model of the non-linearity of the power amplifier, and to determine filter coefficients to be applied by the non-linear filter to the signal components responsively to the inverse model.

10. The apparatus according to claim 9, wherein the adaptation circuit is arranged to filter the samples of the output signal responsively to the inverse model, and is further coupled to compare the intermediate signal to the filtered samples of the output signal in order to determine the filter coefficients.

11. The apparatus according to claim 10, wherein the adaptation circuit is arranged to determine the filter coefficients adaptively using a least mean square (LMS) adaptation process, so as to minimize an error signal based on a comparison of the intermediate signal to the filtered samples of the output signal.

12. The apparatus according to claim 9, wherein the adaptation circuit is further arranged to determine adaptively the input signal components into which the input signal is decomposed by the non-linear filter.

13. The apparatus according to claim 9, and comprising a sampling channel, which couples the output of the power amplifier to the adaptation circuit, wherein the adaptation circuit comprises a channel compensator, which

is adjustable so as to compensate for distortion introduced in the samples of the output signal by the sampling channel.

14. The apparatus according to claim 13, and comprising a calibrator circuit, which is adapted to inject a calibration signal into the sampling channel, so as to determine the distortion introduced in the samples, for use in adjusting the channel compensator.

15. The apparatus according to claim 14, wherein the calibrator circuit comprises a digital logic driver, which generates a square wave of known amplitude and frequency, and a bandpass filter, which filters the square wave so as to provide the calibration signal.

16. The apparatus according to claim 1, wherein the input signal and the intermediate signal are baseband signals, wherein the apparatus comprises an upconverter for upconverting the intermediate signal to a radio frequency (RF) for input to the power amplifier, and wherein the non-linear filter is adapted to apply a low-pass filtering function to the signal components.

17. A method for amplifying an input signal having an input signal power, using a power amplifier characterized by a non-linearity, the method comprising:

decomposing the input signal into a series of input signal components, each such component proportional to a product of the input signal by a respective integer power of the input signal power;

filtering the signal components responsively to the non-linearity so as to generate an intermediate signal for input to the power amplifier; and

amplifying the intermediate signal using the power amplifier.

18. The method according to claim 17, wherein filtering the signal components comprises combining the input signal components so as to generate substantially uncorrelated input signal elements, and filtering each of the substantially uncorrelated input signal elements so as to compensate for the non-linearity.

19. The method according to claim 18, wherein combining the input signal components comprises determining weighting coefficients so as to minimize a correlation between the signal elements, and generating the substantially uncorrelated input signal elements by linear combination of the input signal components using the weighting coefficients.

20. The method according to claim 18, wherein filtering each of the substantially uncorrelated input signal elements comprises decomposing an output signal from the power amplifier so as to generate substantially uncorrelated output signal components, and processing the substantially uncorrelated output signal components in order to determine filter coefficients to be applied in filtering the substantially uncorrelated input signal elements.

21. The method according to claim 20, wherein processing the substantially uncorrelated output signal components comprises filtering the substantially uncorrelated output signal components, and comparing the filtered substantially uncorrelated output signal components to the intermediate signal in order to determine the filter coefficients.

22. The method according to claim 21, wherein comparing the filtered substantially uncorrelated output signal components comprises computing successive differences between the intermediate signal and the filtered substantially uncorrelated output signal components, and adaptively computing the filter coefficients to be applied to each of the substantially uncorrelated input signal components using a respective error signal by the successive differences.

23. The method according to claim 20, wherein combining the input signal components comprises generating the substantially uncorrelated input signal elements by linear combination of the input signal components, and wherein decomposing the output signal comprises adaptively determining weighting coefficients to be applied in generating both the substantially uncorrelated input signal elements and the substantially uncorrelated output signal elements.

24. The method according to claim 18, wherein filtering the signal components comprises summing the filtered substantially uncorrelated input signal elements in order to generate the intermediate signal.

25. The method according to claim 17, wherein filtering the signal components comprises sampling and processing an output signal from the power amplifier so as to produce an inverse model of the non-linearity of the power amplifier, and determining filter coefficients to be applied to the signal components responsively to the inverse model.

26. The method according to claim 25, wherein processing the output signal comprises filtering the sampled output

signal responsively to the inverse model, and wherein determining the filter coefficients comprises comparing the intermediate signal to the filtered, sampled output signal in order to determine the filter coefficients.

27. The method according to claim 26, wherein determining the filter coefficients comprises computing the filter coefficients adaptively using a least mean square (LMS) adaptation process, so as to minimize an error signal based on a comparison of the intermediate signal to the filtered samples of the output signal.

28. The method according to claim 25, wherein decomposing the input signal comprises adaptively selecting the input signal components into which the input signal is to be decomposed.

29. The method according to claim 25, wherein sampling the output signal comprises receiving and processing samples of the output signal via a sampling channel, calibrating the sampling channel so as to determine a distortion introduced by the sampling channel, and correcting the samples so as to compensate for the distortion introduced by the sampling channel.

30. The method according to claim 29, wherein calibrating the sampling channel comprises injecting a calibration signal into the sampling channel, so as to determine the distortion introduced in the samples, for use in correcting the samples.

31. The method according to claim 30, wherein injecting the calibration signal comprises generating a square wave of known amplitude and frequency using a digital logic

driver, and filtering the square wave so as to provide the calibration signal.

32. The method according to claim 17, wherein the input signal and the intermediate signal are baseband signals, wherein amplifying the intermediate signal comprises upconverting the intermediate signal to a radio frequency (RF) for input to the power amplifier, and wherein filtering the signal components comprises applying a low-pass filtering function to the signal components.

33. A receiver, comprising:

a sampling channel, which is adapted to receive and digitize an analog input signal, so as to generate a sequence of digital samples of the input signal;

a calibration circuit, which comprises:

a digital logic device, which is adapted to generate a square wave of known amplitude and frequency; and

a bandpass filter, which is coupled to filter the square wave so as to provide a calibration signal for input to the sampling channel, for use in measuring a distortion introduced by the sampling channel; and

a channel compensator, which is coupled to operate on the digital samples so as to compensate for the distortion measured using the calibration circuit.

34. The receiver according to claim 33, wherein the digital logic device comprises a temperature-compensated emitter-coupled logic (ECL) device.

35. The receiver according to claim 33, wherein the bandpass filter comprises a printed circuit filter.

36. The receiver according to claim 33, wherein the calibration circuit comprises a frequency synthesizer, which is adapted to generate a spectrum of one or more frequency tones for input to the digital logic device, so as to cause the digital logic device to generate the square wave at the known frequency.